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EXAMINER	
CRUTCHFIELD, CHRISTOPHER M	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	**Application No.** 10/532,593	**Applicant(s)** WRAY ET AL.	
	Examiner Christopher Crutchfield	**Art Unit** 2466	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *9/28/2009*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-3 and 7-11* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-3 and 7-11* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. The factual inquiries set forth in *Graham* **v.** *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

   1. Determining the scope and contents of the prior art.
   2. Ascertaining the differences between the prior art and the claims at issue.
   3. Resolving the level of ordinary skill in the pertinent art.
   4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

3. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of 35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

4. **Claims 1 and 10** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Komatsu,* et al. (US Patent No. 6,914,900 B1) in view of *Bao*, et al. (Yin Bao and Adarshpal

Sethi, Performance Driven Adaptive Admission Control for Multimedia Applications, Proceedings of the IEEE 1999 International Conference on Communications, 10 June 1999, Pages 199-203).

Regarding claim 1, *Komatsu* discloses a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls to a second local area network and determining a current packet loss rate based on the packet loss rate of a previous call, Determining said current packet los rate and deciding to drop a call attempt based on the packet loss rate (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of *Komatsu* discloses a system that maintains the packet loss rate from previous calls between two endpoints [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Komatsu fails to disclose a method further comprising using the packet loss rate of multiple previous calls to determine a current packet loss rate based on the packet loss rate of previous calls. In the same field of endeavor, *Bao* discloses a method further comprising using the packet los rate of multiple previous calls to determine a current packet loss rate based on the packet loss rate of previous calls (Pages 199-200 and Fig. 1). (The system of *Bao* discloses

an admission control system that tracks the average of multiple bandwidth probe measurements in order to calculate an admission control threshold [Pages 199-200 and Fig. 1]. The average is prevents the wide distribution of instantaneous values [seen in Fig. 1] from causing the admission control algorithm to oscillate frequently and improperly admit calls based on noise present in the bandwidth sampling process.)

Therefore, since *Bao* suggests the use of multiple sample values to obtain an average value for use in admission control, a person of ordinary skill in the art at the time of the invention would have recognized that the multiple samples of *Bao* could be used in the system of *Komatsu* by averaging the packet loss rate for multiple previous calls in order to provide a more stable measurement for admission control. The motive to combine is to reduce system sensitivity to noise in the drop rate measurement process by averaging multiple drop-rate samples from previous calls.

In the alternative, *Komatsu* could be viewed as teaching a base device comprising an admission control system which uses the packet drop rate of previous calls as a call admission threshold and *Bao* could be viewed as teaching a known technique for improving the measurement of call admission control thresholds comprising the averaging of multiple admission control threshold measurements to reduce the admission control systems sensitivity to noise in the measurement process. Therefore, data averaging of the admission threshold for a call admission controller, as taught by *Bao*, was a known technique for improving call admission control devices at the time of the invention and a person of ordinary skill in the art at the time of the invention would have recognized that this technique could be applied to the similar admission threshold based call admission control system of *Komatsu* by averaging the per call loss based admission thresholds of *Komatsu* over several calls. See KSR Int'l Co. v. Teleflex Inc. 550 U.S. 398, 82 USPQ2d 1385 (Supreme Court 2007).

Regarding claim 10, *Komatsu* discloses a method wherein said step of determining a packet loss rate of a previous call comprises determining the packet loss rate from a first local area network to a second local area network (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (See claim 1, Supra)

Komatsu fails to disclose fails to disclose a method wherein said step of determining a packet loss rate comprises determining the packet loss rate of multiple of previous calls. In the same field of endeavor, *Bao* discloses a method wherein said step of determining a packet loss rate comprises determining the packet loss rate of multiple of previous calls (Pages 199-200 and Fig. 1). (The system of *Bao* discloses an admission control system that tracks the average of multiple bandwidth probe measurements in order to calculate an admission control threshold [Pages 199-200 and Fig. 1]. The average is prevents the wide distribution of instantaneous values [seen in Fig. 1] from causing the admission control algorithm to oscillate frequently and improperly admit calls based on noise present in the sampling process.)

Therefore, since *Bao* suggests the use of multiple sample values to obtain an average value for use in admission control, a person of ordinary skill in the art at the time of the invention would have recognized that the multiple samples of *Bao* could be used in the system of *Komatsu* by averaging the call loss rate for multiple previous calls in order to provide a more stable measurement for admission control. The motive to combine is to reduce system sensitivity to noise in the drop rate measurement process by averaging multiple drop-rate samples from previous calls.

5. **Claim 2** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) in view of *Oran*, et al. (US Pre Grant Publication No. 2006/0034188).

Regarding claim 2, *Odom* discloses a method of call admission control for a continuous stream of data in packet switched networks (Page 1, Second Paragraph) including at least two local area networks (*Odom,* Page 4, Figure 4 and Fourth Paragraph) that communicate with one another across a connecting network (Page 4, Figure 4, WAN) the method comprising the steps of:

a. Determining current packet loss rate for calls from the first local area network to the second local area network (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control.)

b. Deciding to drop the call attempt based on the current packet loss rate (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (See Supra in [a])

c. Transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network [Odom, Page 19, SAA Protocol].)

d. Reflecting the burst of trial data received at the second node back to the first node (Odom, Page 19, SAA Protocol).

e. Receiving the reflected burst of trial data at the first node through the connecting network (Odom, Page 19, SAA Protocol).

f. Comparing the reflected burst of trial data to the transmitted burst of trial data to determine whether transmission of a continuous stream of data can be initiated from the first node in the first local area network to the second node in the second local area network (Odom, Page 19, SAA Protocol, Calculating Planned Impairment Value). (It is inherent that in order to determine packet loss in a ping style test [Odom, Page 18, SAA Probes Versus Pings], the reflected burst of trial data must be analyzed and compared to the data sent to determine if a portion of the burst was lost [i.e. if packet loss occurred].)

g. The burst of trial data comprises a plurality of packets having a size and a priority that corresponds to packets that are to be sent if the call is completed (Page 19, SAA Protocol). (The SAA protocol may also send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Odom fails to disclose the first and second nodes comprise telephones. In the same field of endeavor, *Oran* discloses the first and second nodes comprise telephones and the burst of trial data comprises a plurality of packets having a size that corresponds to packets that art to be sent if the call is completed (Paragraphs 0031-0038 and 0046). (The system of *Oran* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Paragraphs 0031-0038]. The packets may be the same size as the actual media packets that are to follow [Paragraph 0046].)

Therefore, since *Oran* discloses the use of size matching for call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Oran* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data with a priority and size that reflects the size and priority of the following voice packets, as taught by *Oran,* and reflecting the trial burst back to the sender, as taught by *Odom.* The motive to combine is to allow the endpoints to test the connection using a realistic probe packet, lowering the load on the call gateways and increasing the accuracy of the probing by matching the characteristics of the probe packet to the actual media packets that will make up the following voice call.

6. **Claims 3 and 11** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) in view of *Komatsu*, et al. (US Patent No. 6,914,900 B1) and *Zuberi,* et al. (US Patent No. 7,366,097).

Regarding claim 3, *Odom* discloses a method of call admission control for a continuous stream of data in packet switched networks (Page 1, Second Paragraph) including at least two

local area networks (*Odom,* Page 4, Figure 4 and Fourth Paragraph) that communicate with one another across a connecting network (Page 4, Figure 4, WAN) the method comprising:

a. Determining a packet loss rate of previous probes from a first local area network to a second local area network (Odom, Pages 19-24). (The system of *Odam* discloses a SAA system that periodically probes designated endpoints to determine a packet loss rate for connections to the designated endpoints in other LANs and caches the most recent result [Odom, Pages 19-24]. The probes that occurred in the past represent the packet loss rate of previous probes and the current probe represents the current packet loss rate for calls [Odom, Pages 19-24].)

a. Determining current packet loss rate for calls from the first local area network to the second local area network (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network. It then measures the packet loss rate to determine the packet loss rate of calls between the two networks. This value, along with others is used to perform client access control.)

b. Wherein said step of determining a current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of trial data received at the second node back to the first node, receiving the reflected burst of trial data at the first node through the connecting network and comparing the reflected

burst of trial data to the transmitted burst of trial data to determine whether transmission of a continuous stream of data can be initiated from the first node in the first local area network to the second node in the second local area network (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network which reflects the data packets back to the sending gateway for analysis [Odom, Page 19, SAA Protocol].)

c. Wherein the burst of trial data comprises a plurality of packets having a size that corresponds to packets that are to be sent if the call is completed (Page 19, SAA Protocol). (The SAA protocol may also send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set.)

d. Deciding to drop a call attempt based on the current packet loss rate or the success rate of previous probe attempts (Pages 22-24). (The system of *Odom* discloses that when a destination endpoint is cached, the system may periodically probe the endpoint [Pages 22-24]. Connections are then admitted or denied based on both the results of the most current probe and the previous probe [Pages 22-24]. The success or failure of a probe attempt is defined as a probe which meets the minimum Calculated Impairment Factor for a call [Page 19].)

Odom fails to disclose a method further comprising determining the packet loss rate based on data collected from calls and not data probes. In the same field of endeavor, *Komatsu* discloses a method further comprising determining the packet loss rate based on data collected from calls and not data probes (Column 5, Lines 5-20, Column 7, Lines 15-27 and Column 3, Line 24). (The system of *Komatsu* discloses a system that maintains the packet loss rate from a previous calls between two endpoints. [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the aggregated loss statistics for that IP endpoint [i.e. local network] from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24].)

Therefore, since *Komatsu* discloses call admission based on previous call performance, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the historic admission control of *Komatsu* into the teachings of *Odom*. The historic admission control of *Komatsu* can be combined with the system of *Odom* by having the system of *Odam* record connection statistics in its cache, as taught by *Komatsu*, and perform admission control on incoming calls based on current and past call statistics, as taught by *Odom*. The motive to combine is provided by *Komatsu* and is to reduce the delay in setting up a new call (Column 5, Lines 5-20).

Odom as modified by *Komtsu* fails to disclose a method comprising determining the success rate of previous calls from a first local area network to a second local area network and deciding to drop the call attempt based on the current packet loss rate and the success rate of previous calls. In the same field of endeavor, *Zuberi* discloses a method comprising determining a method comprising determining the success rate of previous calls from a first local area network to a second local area network and deciding to drop the call attempt based on the

current packet loss rate and the success rate of previous calls (Column 15, Lines 2-22 and Column 13, Line 30 to Column 14, Line 56). (The system of *Zuberi* discloses a bandwidth assessment probe for assessing bandwidth for transmitting audio [Column 13, Line 30 to Column 14, Line 56 and Column 7, Lines 8-27]. *Zuberi* further discloses that the admission success or rejection of previous connections on the same link is stored in an admission control cache [Column 13, Line 30 to Column 14, Line 56]. When a new connection is made with the system, the system checks the admission control cache, if a previous session on the link was admitted, then the system performs only an abbreviated probe of the stream [Column 15, Lines 2-22 and Column 14, Lines 20-30].)

Therefore, since *Zuberi* discloses using past admission success and present probe results in admitting connections, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the success based probing of *Zuberi* into the system of *Odom* by having the system of *Odom* store the success or failure of previous calls to an endpoint in its cache and to then verify a connection to a cached endpoint using a probe, as taught by *Zuberi*. The motive to combine is provided by *Zuberi* and is to reduce probing traffic connection setup time by requiring only a brief validation of the connection using a shortened probe (Column 15, Lines 2-22).

Regarding claim 11, *Odom* fails to disclose a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network. In the same field of endeavor, *Oran* discloses a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network (Paragraphs 0031-0038 and 0046). (The system of *Oran* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones

[Paragraphs 0031-0038]. The packets may be the same size as the actual media packets that are to follow [Paragraph 0046].)

Therefore, since *Oran* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Oran* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data, as taught by *Oran* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

7. **Claims 7 and 9** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Komatsu*, et al. (US Patent No. 6,914,900 B1) and *Bao*, et al. (Yin Bao and Adarshpal Sethi, Performance Driven Adaptive Admission Control for Multimedia Applications, Proceedings of the IEEE 1999 International Conference on Communications, 10 June 1999, Pages 199-203) as applied to claim 1 and further in view of *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26).

Regarding claim 7, *Komatsu* fails to disclose a method wherein said step of determining said current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network. In the same field of endeavor, *Odom* discloses a method according wherein said step of determining said current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a

second node in the second local area network, reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network which reflects the packets back to the sender to be analyzed [Odom, Page 19, SAA Protocol].)

Therefore, since *Odom* disclose the use of reflected trial data, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the reflected probes of *Odom* into the teachings of *Komatsu* by reflecting probe packets. The motive to combine is to test the connection bi-directionally without having to generate a second set of test packets at the receiving node.

Regarding claim 9, *Komatsu* fails to disclose a method wherein the burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed. In the same field of endeavor, *Odom* discloses a method wherein the burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed (Odom, Page 19, SAA Protocol). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network which reflects the data packets back to the sending gateway for analysis [Odom, Page 19, SAA Protocol].)

Therefore, since *Odam* discloses the use of size and priority matching for call simulation between two endpoints, it would have been obvious to combine the endpoint call simulation of *Odam* with the system of *Komatsu* by having the endpoints transmit a trial burst of data with a priority and size that reflects the size and priority of the following voice packets, as taught by *Odam* and reflecting the trial burst back to the sender, as taught by *Komatsu*. The motive to

combine is to allow the endpoints to test the connection using a realistic probe packet, thereby increasing the accuracy of the probing by matching the characteristics of the probe packet to the actual media packets that will make up the following voice call.

8. **Claim 8** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Komatsu*, et al. (US Patent No. 6,914,900 B1), *Bao*, et al. (Yin Bao and Adarshpal Sethi, Performance Driven Adaptive Admission Control for Multimedia Applications, Proceedings of the IEEE 1999 International Conference on Communications, 10 June 1999, Pages 199-203) and *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) as applied to claim 7 and further in view of *Oran*, et al. (US Pre Grant Publication No. 2006/0034188).

Regarding claim 8, *Komatsu* fails to disclose a method wherein the first node comprises a telephone and said second node comprises a telephone. In the same field of endeavor, *Oran* discloses a method wherein the first node comprises a telephone and said second node comprises a telephone (Paragraphs 0031-0038 and 0046). (The system of *Oran* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Paragraphs 0031-0038]. The packets may be the same size as the actual media packets that are to follow [Paragraph 0046].)

Therefore, since *Oran* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Oran* with the system of *Komatsu* by having the telephone endpoints transmit a trial burst of data, as taught by *Oran* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to

allow the endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

Response to Arguments

1. Applicant's arguments, filed 28 September 2009, have been fully considered but they are not persuasive.

Regarding Applicant's arguments that *Kong* fails to disclose a current packet loss rate based on the packet loss rate of previous calls, and determining the current packet loss rate and deciding to drop a call based on the current packet loss rate, as required by claim 1, applicant's arguments have been rendered moot by the newly supplied reference, *Bao*, et al. (See Claim 1, Supra).

Regarding Applicant's arguments that *Kong* fails to disclose an originating and termination node in a different network, the Applicant's arguments are not found persuasive. Although The Office maintains the position that *Kong*, et al. discloses the use of terminating nodes in different LANs, as the applicant's amendments have eliminated the need to rely on the *Kong* reference in claim 1, this reference has been removed. Instead it is noted that *Komatsu*, et al. also discloses this feature as the monitored call traffic from a packetizing unit flows between the two network endpoints over a router, indicating that the two call endpoints reside in different LANs (See Column 8, Lines 23-30) (See also, Claim 1).

Regarding the remainder of applicant's arguments concerning *Kong,* et al., the arguments are likewise rendered moot by the removal of the *Kong* reference form the rejection.

Regarding Applicant's arguments that *Wing*, et al. (US Patent No. 7496,044) is not a proper prior art reference, the reference to *Wing*, et al. (US Patent No. 7496,044) has been changed to indicate the portions of the pre-grant publication from which it is a continuation (*Odam,* et al., US Pre-Grant Publication No. 2006/0034188), thereby demonstrating a proper prior art date under 35 USC 102(e).

Regarding Applicant's arguments that *Wing/Oran* fails to disclose that the priority of the packets in the burst of trial data corresponds to packets that are to be sent if the call is completed has been considered and is not persuasive. Although the applicant is correct in

stating that the pre-grant publication from which the *Wing* patent gains priority does not have support for matching the priority of probe and voice packet, this feature is also disclosed by *Odon*, et al. (See For Example, Claim 2) (See also Odom, Page 23, SAA Probe Format). Therefore, The Applicant's arguments are found unpersuasive.

Conclusion

2. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL.** See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Christopher Crutchfield whose telephone number is (571) 270-3989. The examiner can normally be reached on Monday through Friday 8:00 AM to 5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Daniel Ryman can be reached on (571) 272-3152. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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